REMARKS

Claims 1-10 were presented for reconsideration and reexamination and, in the aforementioned Office action, were all rejected under 35 U.S.C. §103(a) as allegedly unpatentable over various combinations of cited references. The rejections are respectfully traversed for reasons explained below. Independent claims 1 and 6 have been amended to distinguish the invention more clearly over the cited art. Claims 1-10 remain in the application and are presented for reexamination and reconsideration in light of these remarks.

In sections 2 and 3 of the action, claims 1 and 6 were rejected under 35 U.S.C. §103(a) over Morgan et al. (US 5,329,587) in view of Romesburg (US 5,796,819). It is still Applicant's view that the Examiner's reliance on the Morgan patent is misplaced because the reference is not pertinent to the invention as claimed.

In support of the rejection, the Examiner first asserts: "Microphones positioned to detect speech from a single source and noise is taught by *Morgan et al.* at Figures 8 and 9; col. 7, line 57 - col. 8, line 23." The Examiner further notes: "One of the microphones being designated a reference microphone and other being designated data microphones is taught by *Morgan et al.* at Figures 8 and 9; col. 7, line 57-8 and col. 8, lines 1-23." Further, the Examiner further asserts: "Plurality of bandpass filters for eliminating a known spectral band containing noise is taught by *Morgan et al.* at col. 4, lines27-48."

The reference microphone 48 in Morgan's Figure 8 is positioned close to a primary noise source 50 and generates a reference signal x(t) that is highly correlated with the primary disturbance (noise) to be eliminated. (See column 7, line 64 – column 8, line 1.) It is again noted that Morgan pertains to systems for active noise cancellation or adaptive noise cancellation. (See, e.g., column 1, lines 22-25, and column 2, lines 53-56.) As noted in column 1 of the patent, "(a)ctive noise control in particular involves the generation of a secondary signal (e.g., sound) for the purpose of counteracting the effect of a preexisting noise disturbance." The reference signal x(t) is generated from the microphone 48 in close proximity to the primary disturbance source 50 (i.e., the primary noise source). The principal of active noise cancellation is that the reference signal x(t), which is assumed to be coherent with the noise signal, is used to control a

"secondary control source," a loudspeaker 45 to produce a noise-canceling acoustical signal. The combination of noise from source 50 and canceling "noise" from the loudspeaker 45 results in only a small residual noise, indicated by error signal e(t), within a zone of silence in the vicinity of a microphone 46. Thus, Morgan has only one microphone (46) positioned to detect speech. The other microphone, the "reference" microphone (48), functions only to detect noise from the single noise source (50).

Morgan uses spectral filters to decompose a filtered form y(t) of the noise reference signal x(t) into subbands. The use of spectral or bandpass filters in signal processing is, of course, well known, but clearly Morgan is using bandpass filters in a totally different context from that of the present invention. Morgan is decomposing a noise signal into subbands. In the present invention, bandpass filters are used, one per microphone of an array of microphones, to eliminate from the microphone output signals a known spectral band containing noise.

Therefore, it is clear from Morgan that the system disclosed in Figures 8 and 9: (a) does not disclose an array of microphones, (b) does not disclose use of a reference microphone in the array, but only a "reference" microphone having the sole purpose of producing a signal representative of the primary noise source, and (c) does not disclose the use of bandpass filters that are coupled one to each microphone of an array of microphones.

The Examiner concedes that Morgan does not teach that the signals from the multiple microphones are processed by adaptive filters so that they may be summed together in a signal summation circuit, but further asserts that Romesburg teaches a system for echo cancellation, using two or more microphones, adaptive filtering of the outputs of the microphones and combining the adaptive filtered microphone outputs in a summation circuit. The Examiner concludes that "it would have been obvious ... to modify the system of *Morgan et al.* to implement combining the adaptively filtered outputs of multiple microphones as taught by *Romesburg,* for the purpose of maximizing the speech signal output level and/or minimizing the noise signal output level, as suggested by *Romesburg et al.* (col. 17, lines 13-20)."

Romesburg is concerned with an echo phenomenon encountered with hands-free telephones that utilize a loudspeaker to reproduce a "far-end" speech signal. The

acoustic signal from the loudspeaker will be picked up by a microphone intended for detection of only a "near-end" speech signal, and will be echoed back to the telephone user at the far end of the connection. Romesburg provides a number of adaptive filter configurations intended to cancel the echo signal from the loudspeaker. Romesburg uses adaptive filters, the outputs of which are combined, but there is a clear distinction between the present invention and the Romesburg disclosure. In each of his disclosed embodiments, Romesburg uses the loudspeaker signal as an input to the adaptive filters. See, for example, FIGS. 4-7, 10 and 11, where the loudspeaker signal derived from the phone system on line 10 is applied to update the adaptive filters 14 and 40. The only configuration where Romesburg does not derive filter inputs directly from the phone system line 10 is the one shown in FIG. 2. Here, the filter reference inputs are derived from a "reference" microphone 26, which is described as receiving acoustic signals only from the loudspeaker 20 and a noise source 8. (See column 5, lines 19-21.)

By way of contrast, the reference microphone used in the present invention to provide inputs to the adaptive filters, does not receive loudspeaker signals, but receives acoustic signals both from a speech source and from multiple noise sources. Claims 1 and 6 have been amended to clarify this distinction over the cited art. It is believed that these amendments render the claims more clearly allowable.

The Examiner's proposed combination of the Romesburg and Morgan disclosures makes little or no logical sense. Morgan discloses an active noise cancellation system, in which noise sources are canceled by generating acoustic signals in a loudspeaker, such that the generated signals and the noise signals effectively cancel in a desired zone of silence surrounding a microphone. Neither the Romesburg disclosure, nor the present invention pertains to active noise cancellation systems, and the suggestion that the disclosures of Romesburg and Morgan might be combined has no merit. There is no suggestion in either reference that it might be advantageous to combine them, and one skilled in the art would have no incentive to combine them. Applicant again respectfully urges the Examiner to withdraw Morgan as a reference because it is not relevant to the subject matter of the invention. The

Romesburg disclosure does contain relevant subject matter, but is distinguishable from the invention for the reasons discussed above.

In sections 4 and 5 of the action, claims 2 and 7 were rejected under 35 U.S.C. §103(a) as allegedly unpatentable over Morgan and Romesburg in view of Chance et al. (US 4,584,441). Chance is cited for its disclosure of signal detection to inhibit the adaptation process if voice signals are not detected. Applicant acknowledges that the Chance patent does mention the use of a large signal detector to inhibit operation of an adaptive filter. It is noted, however, that the technique disclosed in Chance is basically the inverse of what is used in the present invention. Chance wishes to inhibit an operation when **a large** signal is detected. In the present invention, a speech detector is employed to inhibit an operation when **no** speech is detected. In any event, claims 2 and 7 are believed to be patentable with the claims from which they depend.

In sections 6-8 of the action, claims 3-5 and 8-10 were rejected under 35 U.S.C. §103(a) as allegedly unpatentable over Morgan and Romesburg in view of Torkkola (US 5,675,659). The latter patent was relied on for its disclosure of speech-conditioning circuitry to reduce reverberation effects. Applicant acknowledges that Torkkola discusses in its background section the desirability of reduction of reverberation, but believes that claims 3 and 8 should be allowable with the claims from which they depend.

In section 8 of the action, the Examiner asserts that the subject matter of claims 4-5 and 9-10 is suggested by Morgan. Specifically, means for filtering data microphone output signals by convolution with a vector of weight values is said to be taught by Morgan at column 5, lines 45-68 to column 6, lines 1-43. As discussed above, Morgan does not include a reference microphone or an array of data microphones. Therefore, Morgan does not suggest many of the other elements of claim 4, such as "means for comparing the filtered data microphone output signals from one of the data microphones with reference microphone output signals and deriving therefrom and error signal." The presence of means for adjusting weight values and fast Fourier transform means in a disclosure pertaining to adaptive filters is not surprising. However, these claims are also believed to be allowable with the claims from which they depend, for reasons discussed above.

For all of the foregoing reasons, the claims as amended are believed to be allowable over the cited art. An action withdrawing the rejections and formally allowing the claims is respectfully requested.

Respectfully submitted,

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ATTACHMENT FOR CLAIM AMENDMENTS
VERSION WITH MARKINGS TO SHOW CHANGES MADE
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1. (Amended) A microphone array processing system for performance enhancement in noisy environments, the system comprising:

a plurality of microphones positioned to detect speech from a single speech source and noise from multiple sources, and to generate corresponding microphone output signals, one of the microphones being designated a reference microphone and the others being designated data microphones, wherein the reference microphone receives acoustic signals both from the speech source and from the multiple noise sources;

a plurality of bandpass filters, one for each microphone, for eliminating from the microphone output signals a known spectral band containing noise;

a plurality of adaptive filters, one for each of the data microphones, for aligning each data microphone output signal with the output signal from the reference microphone; and

a signal summation circuit, for combining the filtered output signals from the microphones, whereby signal components resulting from the speech source combine coherently and signal components resulting from noise combine incoherently, to produce an increased signal-to-noise ratio.

6. (Amended) A method for improving detection of speech signals in noisy environments, the method comprising:

positioning a plurality of microphones to detect speech from a single speech source and noise from multiple sources, one of the microphones being designated a reference microphone and the others being designated data microphones, wherein the reference microphone receives acoustic signals both from the speech source and from the multiple noise sources;

generating microphone output signals in the microphones;

filtering the microphone output signals in a plurality of bandpass filters, one for each microphone, to eliminate from the microphone output signals a known spectral band containing noise;

adaptively filtering the microphone output signals in a plurality of adaptive filters, one for each of the data microphones, and thereby aligning each data microphone output signal with the output signal from the reference microphone; and

combining the adaptively filtered output signals from the microphones in a signal summation circuit, whereby signal components resulting from the speech source combine coherently and signal components resulting from noise combine incoherently, to produce an increased signal-to-noise ratio.